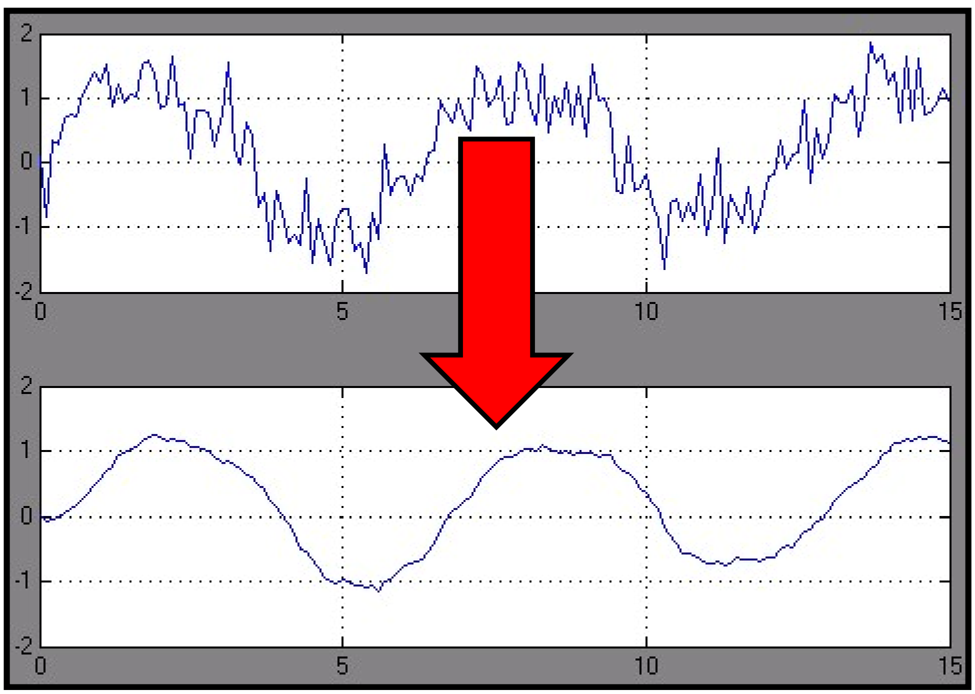
DEPARTMENT OF AEROSPACE ENGINEERING
FACULTY OF ENGINEERING, ARCHITECTURE AND SCIENCE

AER 715 AVIONICS AND SYSTEMS

LABORATORY 1: Introduction to Digital Signal Processing



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DEPARTMENT OF AEROSPACE ENGINEERING
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0. General Safety Rules and Regulations for Laboratories and Research Areas

The following safety rules and regulations are to be followed in all Aerospace Engineering laboratories and research facilities. These rules and regulations are to insure that all personnel working in these laboratories and research areas are protected, and that a safe working environment is maintained.

1. "Horseplay" is hazardous and will not be tolerated.

2. No student may work alone in the laboratory at any time, except to prepare operating procedures for equipment or data write-up/reduction/simulations.

3. Required personal protective equipment (PPE) will be provided by the Department for use whenever specified by the Faculty, Engineering Support or Graduate/Teaching Assistant, i.e., hearing protection, face shields, dust masks, gloves, etc.

4. Contact lenses will not be worn in the laboratory when vapours or fumes are present.

5. Safety glasses with side shields and plastic lenses will be required when operating targeted class experiments as outlined in the experimental procedures. Splash goggles or face shields will also be provided and worn also, for those experiments which have been identified as a requirement.

6. Each student must know where the location of the First Aid box, emergency equipment, eye wash station is, if required in the laboratories, shops, and storage areas.

7. All Faculty, Engineering Support and Graduate/Teaching Assistants must know how to use the emergency equipment and have the knowledge to take action when an accident has occurred, i.e., emergency telephone number, location, emergency response services.

8. All Faculty, Engineering Support and Graduate/Teaching Assistants, and Research Assistants, must be familiar with all elements of fire safety: alarm, evacuation and assembly, fire containment and suppression, rescue.

9. Ungrounded wiring and two-wire extension cords are prohibited. Worn or frayed extension cords or those with broken connections or exposed wiring must not be used. All electrical devices must be grounded before they are turned on.

10. All Faculty, Engineering Support and Graduate/Teaching Assistants, and Research Assistants, must be familiar with an approved emergency shutdown procedure before initiating any experiment.

11. There will be NO deviation from approved equipment operating procedures.

12. All laboratory aisles and exits must remain clear and unblocked.

13. No student may sniff, breathe, or inhale any gas or vapour used or produced in any experiment.

14. All containers must be labeled as to the content, composition, and appropriate hazard warning: flammable, explosive, toxic, etc.

15. The instructions on all warning signs must be read and obeyed in all laboratories and research facilities.

16. All liquid and solid waste must be segregated for disposal according to Faculty, Engineering Support or Graduate/Teaching Assistant instructions. All acidic and alkaline waste should be neutralized prior to disposal. NOTE: NO organic waste material is to be poured down the sink or floor drains. These wastes should be property placed in designed waste disposal containers, labeled and stored in the department's flammable storage cabinet which is ventilated and secured.

17. Good housekeeping must be practiced in all teaching and research laboratories, shops, and storage areas.

18. Eating, drinking, tobacco products, gum chewing or application of makeup is strictly prohibited in the laboratories and shops.

19. Only chemicals may be placed in the "Chemicals Only" refrigerator. Only food items may be placed in the Food Only refrigerator. Ice from any refrigerator is not be used for human consumption or to cool any food or drink.

20. Glassware breakage must be disposed in the cardboard boxes marked "Glass Disposal". Any glassware breakage and malfunctioning instruments or equipment must be reported to the Faculty, Engineering Support or Graduate/Teaching Assistant present.

21. All injuries, accidents, and "near misses" must be reported to the Faculty, Engineering Support or Graduate/Teaching Assistant. The Accident Report must be completed as soon as possible after the event by the Faculty, Engineering Support or Graduate/Teaching Assistant and reported to the Departmental Safety Officer immediately. Any person involved in an accident must be sent or escorted to the University Health Centre. All accidents are to be REPORTED.

22. All chemical spills are to be reported to the Faculty, Engineering Support or Graduate/Teaching Assistant, whose direction must be followed for containment and cleanup. Faculty, Engineering Support or Graduate/Teaching Assistant will follow the prescribed instructions for cleanup and decontamination of the spill area. The Departmental Safety Officer must be notified when a major spill has been reported.

23. All students and Faculty, Engineering Support or Graduate/Teaching Assistant must wash their hands before leaving targeted laboratories, research facilities or shops.

24. No tools, supplies, or any other items may be tossed from one person to another.

25. Compressed gas cylinders must be secured at all times. Proper safety procedures must be followed when moving compressed gas cylinders. Cylinders not in use must be capped.

26. Only gauges that are marked "Use no oil" are for Oxygen cylinders. Do not use an oiled gauge for any oxidizing or reactive gas.

27. Students are never to play with compressed gas hoses or lines or point their discharges at any person.

28. Do not use adapters or try to modify any gas regulator or connection.

29. There will be no open flames or heating elements used when volatile chemicals are exposed to the air.

30. Any toxic chemicals will be only be exposed to the air in a properly ventilated Fume Hood. Flammable chemicals will be exposed to the air only under a properly ventilated hood or in an area which is adequately ventilated.

31. Personal items brought into the laboratory or research facility must be limited to those things necessary for the experiment and safe operation of the equipment in the laboratories and research facilities.

32. General laboratory coats, safety footwear are not provided by the Department of Aerospace Engineering, although some targeted laboratories and research areas will be supported by a reasonable stock of protective clothing and accessories, i.e., gloves, welding aprons, dust masks, face shields, safety glasses, etc.

33. Equipment that has been deemed unsafe must be tagged and locked out of service by the Technical Officer in charge of the laboratory or research facility. The Departmental Safety Officer must be notified of the equipment lockout IMMEDIATELY!

34. In June 1987 both the Federal & Ontario Governments passed legislation to implement the workplace hazardous material information system or WHMIS across Canada. WHMIS was designed to give workers the right-to-know about hazardous material to which they are exposed to on the job. Any person who is required to handle any hazardous material covered by this act should first read the label and the product's material safety data sheet (MSDS). No student is to handle any hazardous materials unless supervised by a Faculty, Engineering Support or Graduate/Teaching Assistant. The laboratory Technical Officer, Faculty, Engineering Support or Graduate/Teaching Assistant is responsible for ensuring that any hazardous materials are stored safely using WHMIS recommended methods and storage procedures. All MSDS must be displayed and stored in a readily accessible place known to all users in the workplace and laboratory

35. All the foregoing rules and regulations are in addition to the Occupational Health and Safety Act, 1987.

36. Casual visitors to the laboratory and research areas are to be discouraged and must have permission from the Faculty, Engineering Support or Graduate/Teaching Assistant to enter. All visitors must adhere to the safety guidelines and is the responsibility of the visitor.

37. Only the Safety Officer may make changes to these policies upon confirmation of the Safety Committee and approval of the Department Chair.

TABLE OF CONTENTS

1. Instructions

- **SAFETY FIRST – DO NOT PUT YOUR FINGERS OR ANY LOOSE ITEMS IN THE SERVOMOTOR GEARS.**
- Download the lab manual and associated files from D2L and save it on the PC computer in a folder called LAB1.
- Read the instructions in the laboratory manual carefully and follow the specified procedures. This lab is to be done with **two (2)** students.
- Answer all questions designated by the symbol.
- At the end of the lab, submit one lab worksheet along with the standard Ryerson Aerospace Assignment/Laboratory Cover Sheet. Each student must attend the laboratory and sign the Cover Sheet in order to receive a mark.

2. Digital Signal Processing

2.1 Purpose

The objective of this lab is to introduce to the student the basics of digital signal processing (DSP) and digital filtering. At the end of this laboratory, the student should have learned the following:

- what digital signal processing is;
- the basics of a finite impulse response (FIR) filter;
- simulation of a moving average filter in Simulink;
- application of a moving average filter to the signal output from an angular rate sensor (gyro); and
- comparison of different filters.

2.2 Apparatus

To complete this lab, the following hardware is required:

- Quanser UPM 1503 or VoltPAQ-X1 Power Module;
- Quanser Q2-USB, Q4, or Q8 PCI Board;
- Quanser SRVO2-ET servomotor plant; and
- PC equipped with MATLAB/Simulink software.

2.3 Introduction to Digital Signal Processing (DSP)

The subject of DSP is vast and is usually taught as an entire course in engineering. The purpose of this laboratory is to provide the student with a brief introduction to DSP and its uses. The student is encouraged to read the materials listed in the references section to get a more comprehensive coverage of the subject.

2.3.1 What is DSP?

Digital signal processing is literally the processing of signals through digital means. Generally, the signals we are analyzing are from sensors which use continuous analog voltages to represent the process they are being used to measure. We would like to use these signals to obtain some meaningful information about the underlying process that we are measuring. DSP allows us to take these analog signals, convert them to digital form and easily manipulate them. The signals processed in this manner are a sequence of numbers that represent samples of a continuous variable in a domain such as time, space, or frequency.

2.3.2 Why DSP?

DSP is an extremely powerful tool that has allowed us to make huge technological advancements in many science and engineering fields. A typical example of the use of DSP is in radar (radio detection and ranging). A simple radar system uses a radio transmitter with a directional antenna to send out high energy pulses. These radio waves reflect back off of the moving aircraft and are detected by a receiving antenna near the transmission site. The distance to the object is calculated by determining the elapsed time between the transmitted and received pulse (time of flight, TOF). Modern radar has been revolutionized through the advent of DSP and has been modernized in three particular areas: 1) DSP compresses the pulse after receiving it, improving distance measurement without affecting operating range; 2) DSP can filter the received signal to decrease noise; and 3) DSP enables on the fly changes of pulse shape and length. Nowadays, DSP has become one of the most powerful technologies that will shape science and engineering in the twenty-first century.

2.3.3 Converting Signals from Analog to Digital

Most DSP applications deal with analog signals. Before we can conduct DSP on these signals, they need to be converted from analog (continuous) to a digital (discrete) representation. This conversion process is accomplished using a special electronic device named analog to digital converter (ADC). There are two important aspects to a digital signal, namely:

1. Sampling – converts the time variable from continuous to discrete
2. Quantization – converts the measured variable (usually voltage) from continuous to discrete.

Both of these aspects restrict how much information a digital signal can contain. Figure 1.1 below shows a typical analog to digital conversion process.

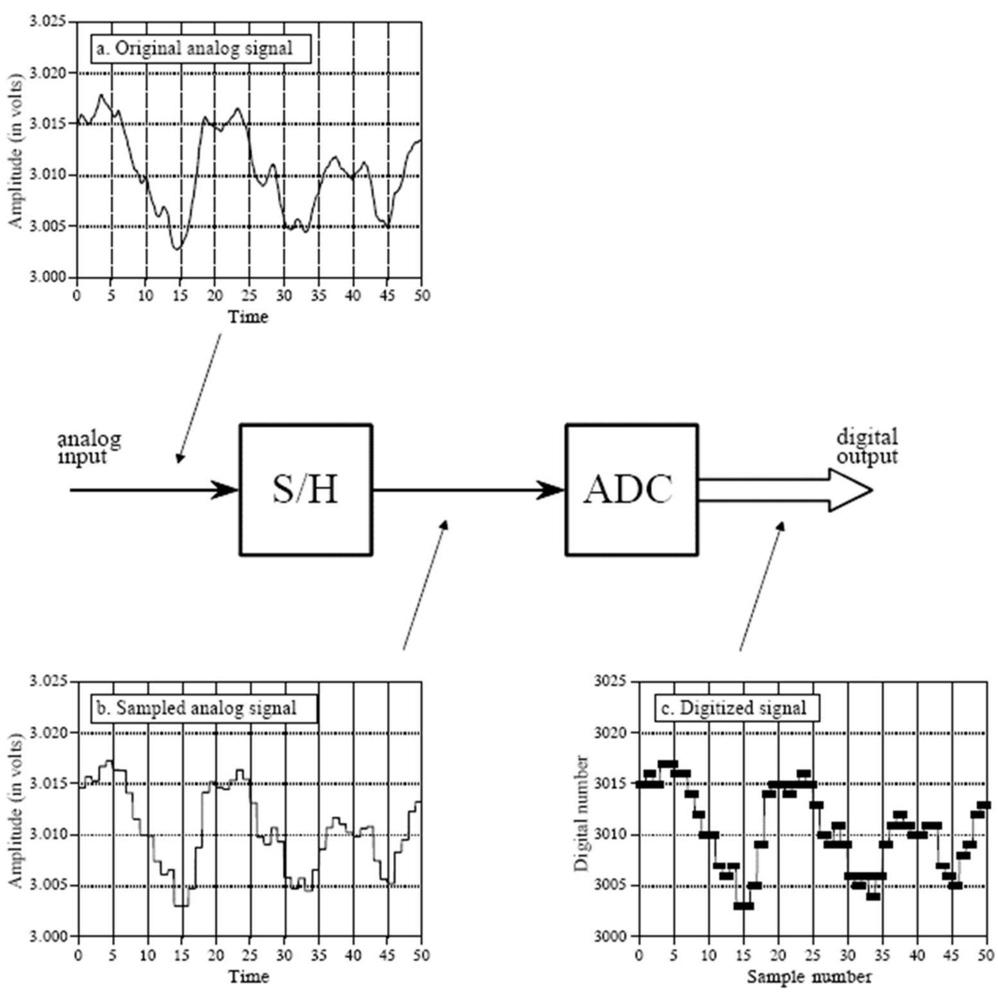


Figure 1.1 A waveform is shown undergoing the digitization process. The process is split into two stages to show the effect of sampling and quantization separately. The first stage is the sample and hold (S/H) where the instantaneous value of the signal is recorded when the periodic sampling takes place. In the second stage,

the ADC converts the voltage to the nearest integer number (in this case each voltage is converted to an integer number is the range of 0 to 4095 or a 12-bit conversion). (Smith, 1999, p.37)

Of **critical importance** is the knowledge that when a signal is converted from analog to digital form, some information could be lost during the conversion process. This information loss is due to factors known as quantization errors such as:

- inaccuracies in the measurement;
- uncertainty in timing; and
- limit on the duration of the measurement.

Analog signals are usually sampled at **fixed** intervals of time. The signal is timed against a clock and the value within that interval is held until the next interval time at which it is then recorded. This produces a quantization error, since it is possible that the voltage measurement during the interval may fluctuate. In addition, the digital number may not be an exact representation of the voltage value. For example, if we consider a 12-bit ADC and assume that our measuring range is from 0 to 4.095 V, the minimum resolution that can be measured is 1 mV (4.095/4095 = 0.001 V). So if we had a truly accurate measurement of 3.0015 V this value has to be stored as integer of either 3001 or 3002, since there are not enough bits to store the fourth decimal value.

2.3.4 The Nyquist Theorem

An important theory in digital signal processing is the Nyquist Theorem. This theorem points out that a continuous signal can only be sufficiently sampled, if it does not contain frequencies above one-half of the sampling rate. For example, if a signal is sampled at a rate of 1000 samples per second, we would need the analog signal to be comprised of frequencies below 500 Hz. If there are frequencies above this limit, they will be aliased to frequencies between 0 and 500 Hz, merging with whatever information was legitimately contained in the signal. Aliasing is the phenomenon of sinusoids changing frequency (and/or phase) during sampling. Aliasing causes corruption of the signal and as such the original signal cannot be reconstructed from the samples. This is illustrated in Figure 1.2(d) below. Preventing aliasing is an important aspect of ADC; however we will not go into any further details. Additional information is available in reference [1].

2.3.5 Digital Filters

Digital filters are extremely important in DSP. Digital filters have two specific purposes. To:

1. Separate combined signals
2. Restore distorted signals.

Analog (or electronic) filters can be used for these same tasks; however digital filters are far superior in their performance capabilities.

Every linear filter has a special response to an impulse, step, or frequency input. It is the resulting behaviour of the filter's response from these three inputs that determines the performance of the filter. If any one of the three responses is fixed, the other two can be determined; however, there exists a tradeoff since we cannot design a filter to work optimally for more than one type of input response.

There are two classes of digital filters: convolution (finite impulse response - FIR) and recursion (infinite impulse response - IIR), as shown in Figure 1.3. We will get into more detail regarding the moving average and finite impulse response (FIR) filter. An FIR filter implies that it has a finite number of non-zero values in its output response. This is contrary to an infinite impulse response (IIR) filter which has an infinite number of non-zero output response values (e.g., a filter with a decaying exponential response).

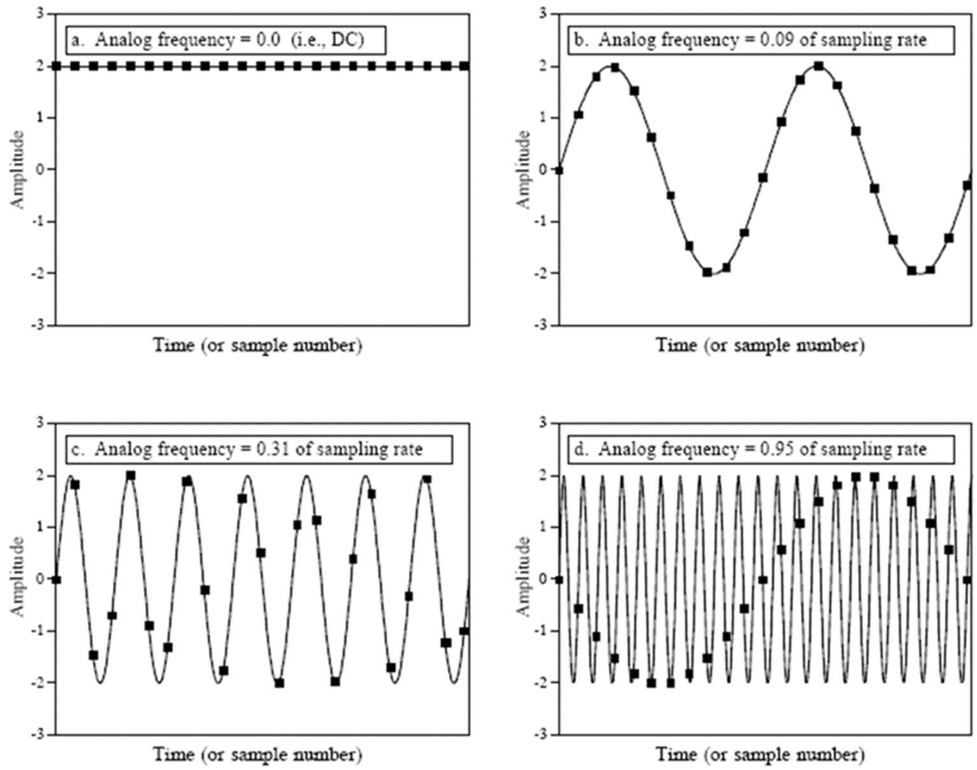


Figure 1.2 The above images illustrate examples of proper and improper sampling of a signal. Figures (a), (b), and (c) show proper sampling of sinusoidal waves; each continuous signal forms a unique one-to-one pair with its pattern of samples. However, in (d) the frequency of the sine wave is greater than the Nyquist frequency which results in aliasing (Smith, 1999, p.41).

FILTER USED FOR:	FILTER IMPLEMENTED BY: Convolution *Finite Impulse Response (FIR)*	Recursion *Infinite Impulse Response (IIR)*
Time Domain *(smoothing, DC removal)*	Moving average	Single pole
Frequency Domain *(separating frequencies)*	Windowed-sinc	Chebyshev
Custom *(Deconvolution)*	FIR custom	Iterative design

Figure 1.3 Filter classification divided by use and implementation (Smith, 1999, p.275)

2.3.6 Filter Transfer Function

Mathematically, the output response of a linear n^{th} order FIR filter, having an impulse response $h(n)$ to an arbitrary input $x(n)$, is defined by the linear convolution sum as:

$$y(n) = \sum_{k=0}^{n-1} h(k)x(n-k) = \sum_{k=0}^{n-1} h(n-k)x(k)$$

That was a mouthful… the convolution is simply a mathematical operator that takes two functions, in this case *h* and *x*, and produces a third function *y*. The function *y* represents the amount of overlap or blending between *h* and *x*.

In the *z*-transform domain, the filter's transfer function is:

$$H(z) = \sum_{k=0}^{n-1} h_k z^{-k}$$

What is the *z*-transform you may ask? The *z*-transform is analogous to the Laplace Transform in continuous signals; it is represented as:

$$z \triangleq e^{sT_s} \qquad \text{or} \qquad z^{-1} \triangleq e^{-sT_s}$$

where *s* is the Laplace operator and T_s is the sampling time. You can look up the *z*-transform on Wikipedia for more details.

An aside… Analog filters, constructed of resistors, capacitor, and inductors can also be represented in DSP. These classic filters are the antithesis of FIR filters and are called IIR (infinite impulse response) filters. The transfer function of an IIR filter in the *z*-domain is given as follows:

$$H(z) = Kz^{n-m} \frac{\prod_{i=0}^{m-1}(z - z_i)}{\prod_{i=0}^{n-1}(z - p_i)} = \frac{N(z)}{D(z)}$$

The IIR filter contains both zeros (numerator) and poles (denominator) compared to the FIR which consists of only zeros. The poles give rise to the filter having an output response that can persist indefinitely due to feedback. IIR filters are advantageous in applications requiring strict frequency selection and low filter order (the number of poles/zeros).

2.3.7 The Moving Average Filter

The moving average (MA) filter is the easiest type of filter to understand. This also makes it one of the most common filters used in DSP. The MA filter is especially suited for noise suppression. It has characteristics of a low pass filter (i.e., passes low frequency signals with minimal amplitude attenuation) and has optimal step response performance; however, its frequency response is often limited.

The MA filter operates by averaging a number of input values to produce each point in the output signal. The equation for the MA filter is as follows:

$$y(n) = \frac{1}{M} \sum_{k=0}^{M-1} x(n-k)$$

where *x* is the input signal, *y* is the output signal, and *M* is the number of points in the average. The filter order is one less than the number of points (or taps). In the subsequent sections we will use Simulink to test the moving average filter before applying the filter to a real-world signal.

Complete the following two warm-up exercises:

1. Write down the equation (symbolically) for the 40^{th} output for a 4-point (3^{rd} order) moving average filter. Use the equation in Section 2.3.7.
2. The analog to digital converter (ADC) on the Q4 data acquisition and control board has a resolution of 14-bits. Determine the smallest voltage that can be resolved by this ADC? The full-scale voltage range is 20V (bipolar ±10V). **Hint:** See Section 2.3.3.

3. Part A: Simulation Exercises (Moving Average Filter)

Start **MATLAB** and launch Simulink from either the icon on the toolbar, or by typing *simulink* at the command prompt. Download and open the Simulink file "***Filter_Simulation.slx***" contained on D2L under the AER715 course tab.

The model is as shown in Figure 1.4 below. The model contains a sine wave that is randomly infused with noise (random number). We will pass the noisy signal through three parallel filters and observe the output results.

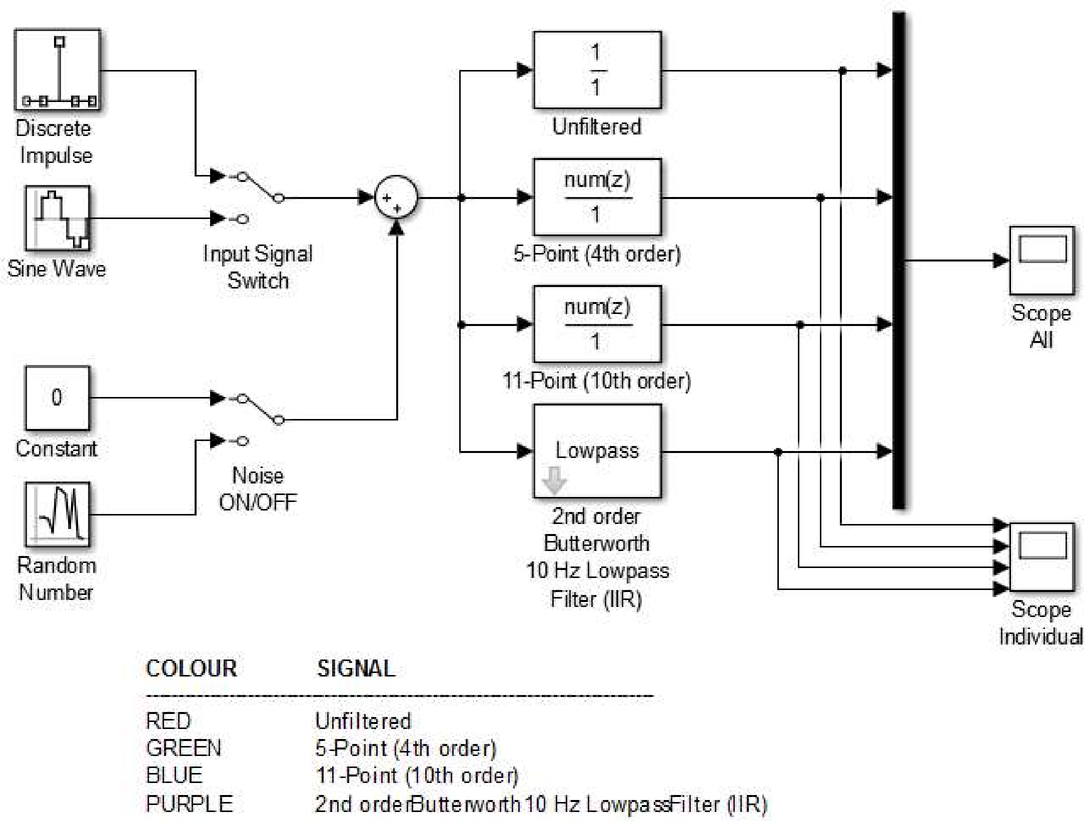
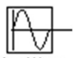
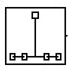
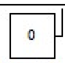
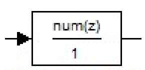
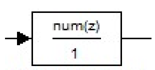


Figure 1.4 Simulink model using a FIR moving average (MA) filter

Set the following parameters for the individual Simulink blocks by double clicking on them.

Block	Parameter Values
 Sine Wave	Amplitude = 4 Bias = 0 Frequency = 2*pi Phase = 0
 Random Number	Mean = 0 Variance = 0.2 Initial Seed = 111 Sample Time = -1
 Discrete Impulse	Delay = 100 (starts impulse at 1 second) Sample time = 0.01 (Sampling Frequency = 100 Hz) Samples per frame = 1
 Constant	Constant value = 0 Sample time = inf
 5-Point (4th order)	Num = 1/5*ones(1,5,'double') Den = [1] Sample Time = -1
 11-Point (10th order)	Num = 1/11*ones(1,11,'double') Den = [1] Sample Time = -1
 2nd order Butterworth 10 Hz Lowpass Filter (IIR)	Impulse Response: IIR Order: 2 Frequency Constraints: 3dB point F3dB: 10 Hz Design method: Butterworth

Note: The command “ones” creates an array of ones.

Set the **simulation time** to **4 seconds** and make sure the **simulation clock** is set to "**Normal**" (See the arrow in Figure 1.4).
Next go to the main toolbar and set the **step size**: **Simulation >> Configuration Parameters >> Fixed-Step Size = 0.01**

Note: Sampling Rate, F_s = 1/Step Size = 100 Samples/s = 100 Hz

Open the “**Scope All**” and “**Scope Individual**” blocks.

Make sure the “**Input Signal Switch**” and “**Noise ON/OFF**” switch are set to “**<u>Discrete Impulse</u>**” and “**<u>Constant</u>**” respectively. Double click on the switches if you need to flip them. Next, select ▶ to run the simulation.

★3.
Look at the signals in the scopes and write down your observations with respect to the impulse response of all the filters. Do the filters behave as expected?

Now flip the "**Input Signal Switch**" to the "**Sine Wave**" by double clicking it. Next, select ▶ to run the simulation.

★4.
The time delay of a FIR filter is given by the following simple formula:

$$t_d = \frac{n-1}{2 \cdot F_s}$$

Compute the time delay for the two moving average filters. Use the "Scope all" graphs to verify your results. Zoom 🔍 into the peak of the sine wave to help you. Are the results the same?

What is the delay of the 2nd order Butterworth filter (measure on the graph)?

Now flip the "**Noise ON/OFF**" switch to the "**Random Number**" block by double clicking it. Next, select ▶ to run the simulation.

★5.
Look at the noisy sine wave input and filtered signals in the four scopes and write down your observations on the worksheet. **Hint:** Figure 1.5 below may help you to answer Question 5.

★6.
Now set the frequency of the sine wave equal to 4*pi (2Hz) and observe the new result. Repeat for a frequency of 8*pi (4Hz) and 20*pi (10Hz). What is happening to the signals? What does this tell us about the limits of the moving average filter? What about the IIR Butterworth Filter? **Hint:** See Section 2.3.6.

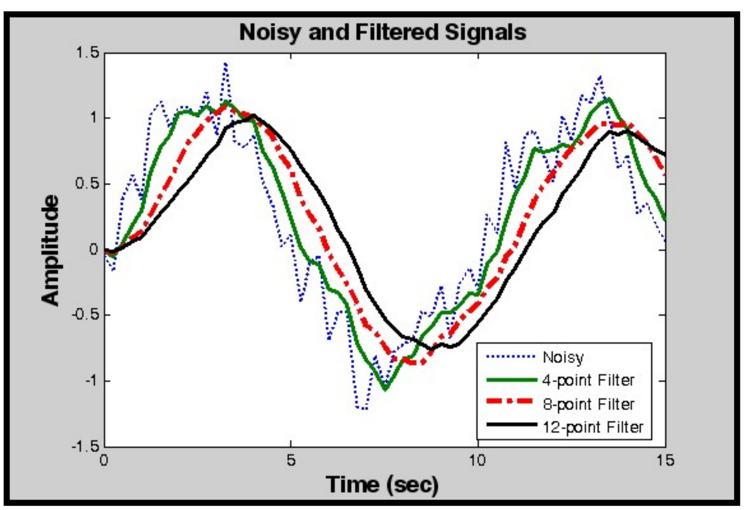


Figure 1.5 Example of an unfiltered and moving average filtered signals

4. Part B: Laboratory Exercises (Implementing the Moving Average Filter)

We are now going to filter the data obtained from a real sensor using the same filters used in the simulation. In particular we are going to look at the output of a tachometer mounted on the end of a DC motor. This sensor is able to measure angular velocity at up to 5000 RPM, with a sensitivity of 1.5 mV/RPM +/- 0.2%. Additional details regarding this sensor are available in the Appendix.

4.1 Part B-1

Download from D2L and open up the Simulink file "***Tachometer_Q#.slx***" that pertains to you HIL board (ask the GA if you are unsure). The model is shown in Figure 1.6 below.

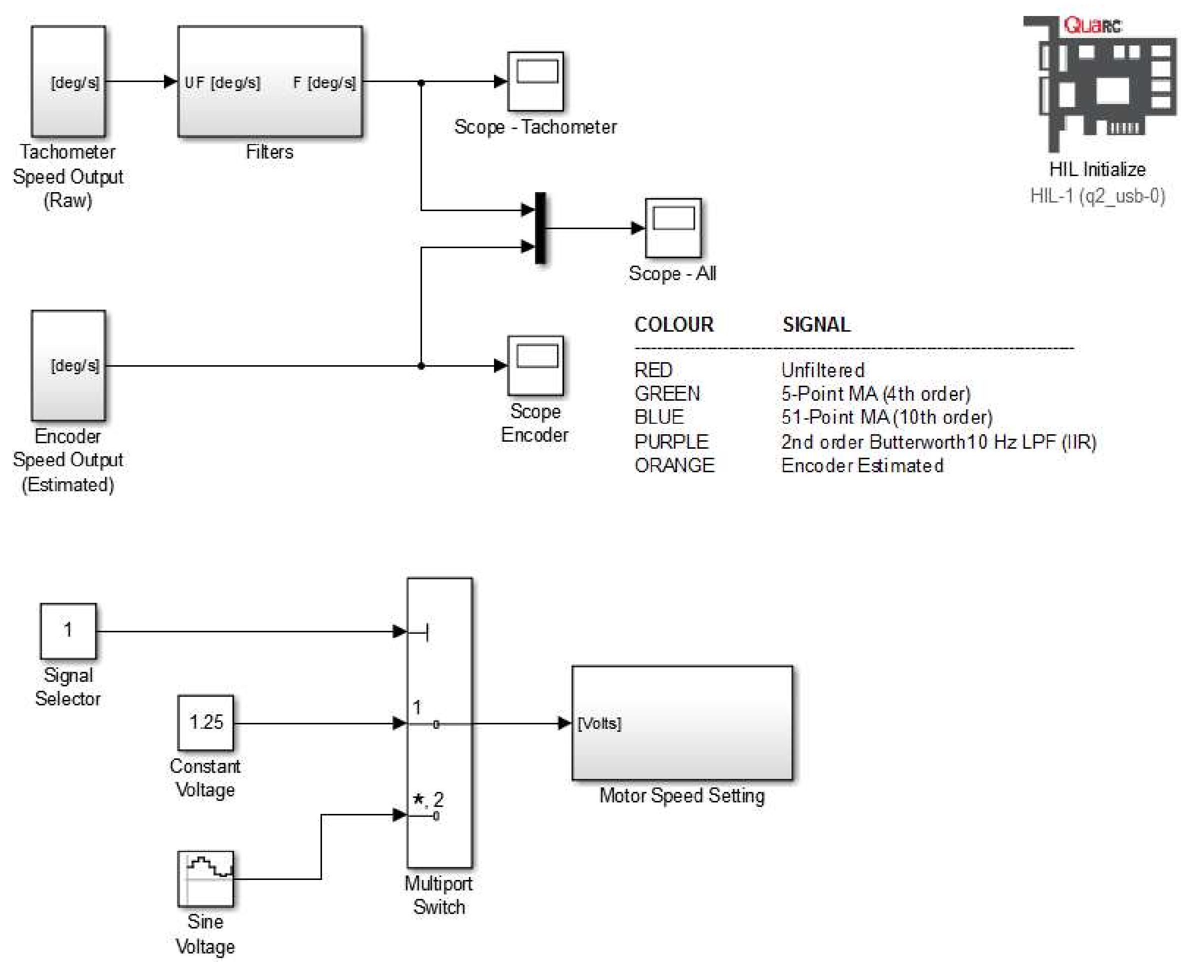
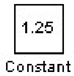
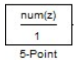
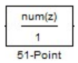
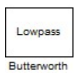


Figure 1.6 Simulink model to measure tachometer signals from various voltage motor inputs

In this first exercise we will operate the motor at a constant voltage. We will assume that the motor will turn at a constant rate (open loop control, no feedback). Adjust the parameter values of the blocks to those in the following table:

Block	Parameter Values
	Constant value = 1.25 Sample Time = inf
	Num = 1/5*ones(1,5,'double') Den = [1] Sample Time = -1
	Num = 1/51*ones(1,51,'double') Den = [1] Sample Time = -1
	Impulse Response: IIR Order: 2 Frequency Constraints: 3dB point F3dB: 10 Hz Design method: Butterworth

Note 1: The individual filters are located in the "Filters" subsystem block.
Note 2: The "-1" Sample Time indicates that the sample time is inherited from the settings in the Configuration Parameters menu or the previous block.
Note 3: The system sampling rate for the following hardware experiment is 200 Hz.

Ensure the following settings are made before proceeding with the test run:

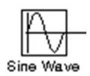
1. Set the "**Signal Selector**" block value to **1**
2. **Simulation >> Model Configuration Parameters >> Additional Options >> Fixed step size = 0.005**
3. **QUARC >> Build** (converts model to real-time c code)

Open the "**Scope – All**" block and the individual scope blocks located in the Filters subsystem block. Next, select **QUARC >> Start** to run the simulation.

★7. What do you see? How does the accuracy of the *unfiltered* tachometer compare to the encoder speed values? Write your observations on your worksheet.
★8. Assuming the sensor is operating within specifications, what factors could be contributing to the variations (noise) in the signal? ***Hint:*** *Press the stop icon and use the zoom functions on the scope toolbar to take your measurements?*
★9. Using the Nyquist theorem in Section 2.3.4, what is the maximum frequency that our system sampling rate allows us to ascertain from the sampled signal?

4.2 Part B-2

Adjust the parameter values of the blocks to those in the following table:

Block	Parameter Values
	Amplitude = 0.25 Bias = 1 Frequency = 2*pi Phase = 0

Complete the following step before proceeding with the test run:

1. Set the "**Signal Selector**" block value to **2**
2. **QUARC >> Build** (converts model to real-time c code)

Open the "**Scope – All**" block and the individual scope blocks located in the Filters subsystem block. Next, select **QUARC >> Start** to run the simulation.

10.
Record your observations with respect to the relative performance of the filters.

11.
Print a copy of the "Scope All" figure. Make sure to label the individual signals on your graph.

5. References

[1] Smith, S.W. "The Scientist and Engineer's Guide to Digital Signal Processing." 2nd Ed. California Technical Publishing, San Diego, California, 1999.

[2] The BORES Signal Processing DSP course - Introduction to DSP. Bores Signal Processing. http://www.bores.com/courses/intro/index.htm. Last updated: 10/14/2014. Last visited: 08/22/2017.

6. Appendix

Series 2251 … S

See beginning of the Section for Ordering Information

Characteristics of the DC-Motor-Tacho Combination

Series	mechanical time constant τ_m	moment of inertia J	angular acceleration $\alpha_{max.}$	frequency	weight response
2251 U 4.5 S 1.5 G	16 ms	$4.93 \cdot 10^{-5}$ oz-in-sec²	$52 \cdot 10^{3}$ rad s⁻²	1,500 Hz	3.21 oz
2251 U 006 S 1.5 G	15 ms	$3.68 \cdot 10^{-5}$ oz-in-sec²	$56 \cdot 10^{3}$ rad s⁻²	1,500 Hz	3.21 oz
2251 U 012 S 1.5 G	16 ms	$4.12 \cdot 10^{-5}$ oz-in-sec²	$56 \cdot 10^{3}$ rad s⁻²	1,500 Hz	3.21 oz
2251 U 024 S 1.5 G	15 ms	$2.65 \cdot 10^{-5}$ oz-in-sec²	$59 \cdot 10^{3}$ rad s⁻²	1,500 Hz	3.21 oz

The characteristics of the **DC-Micromotor Series 2233 … S**, can be veiwed on the corresponding motor specification page.

Tachogenerator		1.5 G	
EMF constant	K_E	1.5	mV/rpm
		14.325	mV/rad s⁻¹
Tolerance of EMF constant		± 2	%
Load resistance	R_L	≥ 25	kΩ
Operating speed, max. continuous	$n_{e\,max}$	≤ 5,000	rpm
Terminal resistance	R	260	Ω
Ripple, peak-peak, typical		7	%
Ripple frequency, cycles		14	per turn
Linearity, without load, between 500 and 5,000 rpm		± 0.2	%
Reversion error		± 0.2	%
Temperature coefficient of EMF		0.02	% / °C
Temperature coefficient of armature resistance		0.4	% / °C
Rotor inductance	L	3,000	µH
Direction of rotation		reversible	
Polarity		dependent on direction of rotation	